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Description

The present invention refers to a programmable signal processing device, mainly intended for hearing aids, and of the kind which includes an electronically controlled signal processor.

Background of the Invention

Impaired hearing is today a very common handicap. It is above all, elderly people and people who are exposed to loud noise, that are affected. We do not discuss the causes of impairments in detail here, but only note that today it is practically impossible to treat these impairments in a medical way. The most common method today to re-establish, at least partly, the hearing of the affected patient, is to let the patient use some type of hearing aid. High demands must be put on such hearing aids, i.e. their frequency response must be adjusted to the patients hearing deficiency and it must also be possible to amplify desired sounds as for example normal conversation. To suit all normally occurring environmental situations it is not unusual that the same patient today has two or more hearing aids, which he or she alters between. The hearing aids must also be small and convenient to use.

Today there exist about a hundred different types of hearing aids on the market and it is therefore difficult for the person responsible for the fitting to decide which one is optimal in the individual case.

An estimate is that one out of four hearing aids is not acceptable by the patient and therefore the hearing aid is not used. As about 2.3 million hearing aids (1980) are distributed in the world every year, there is a great need for improving the devices and to develop more accurate and simplified fitting methods.

It would also be desirable to reduce the number of hearing aid types on the market to a few main types on condition that these main types can be adapted to each individual need.

Different types of filters with variable frequency response are earlier disclosed in the patent literature. Such filters are for example disclosed in the US Patent Publication No. 3,989,904 filed Dec. 30, 1974, with the title "Method and apparatus for setting an aural prosthesis to provide specific auditory deficiency corrections", and in the Danish Patent Publication No. 138,149, filed Feb. 23, 1973, with the title "Kobling til brug i et høreapparat og i et apparat til måling af menneskelige høredefekter".

The American invention refers to a device intended for adjusting a hearing aid in such a way that the gain in different frequency bands and maximum power output can be adjusted at the fitting procedure. The device has a number of disadvantages. For example the hearing aid can be optimally adjusted for only one sound environment.

The Danish invention refers to a similar device where every filter individually can be adjusted with respect to the amplification. In this invention

only one frequency response can be set and the patient can hear well only optimally in just one sound environment, for example at normal conversations at home, while the device can be practically impossible to use in other sound environments, such as for example at place of work with disturbing background noise, traffic environment or at meetings, parties and the like.

In the U.S. Pat. No. 4,185,168 is also disclosed a method of and means for filtering near-stationary, relatively long duration noise from an input signal containing information such as speech or music. The invention is mainly a noise analyzer and adapted to automatically adjust band pass filters. The device is arranged to reduce the signal-to-noise ratio and is not aimed for different listening situations. The present invention makes it possible to automatically or manually change the parameters according to the acoustical environment and the preprogrammed hearing loss of the person who is wearing the device.

The precharacterising part of claim 1 is based on this document.

In the US patent 4,187,413 is further disclosed a hearing aid which includes a memory multiplexer for loading of multiplier coefficients for adapting the transfer function to different types of hearing deficiency. The hearing aid is possible to reprogram without disassembly. The programmed parameters are however related to one present hearing deficiency and not to various listening situations which can occur. i.e. only one signal process can be programmed at one time. There is therefore no possibility to alter between a number of different signal processes suitable for various sound environments.

The Objects of the Invention

An object of the invention is to provide a programmable signal processing device which automatically, or controlled by the user, select the signal process, which is best suited to the particular sound environment. Further objects of the invention are that the signal processing device should be easy to use and comfortable to wear for the person with impaired hearing, easy to adjust/program and cheap to produce.

By means of such a signal processing device the following functions among others could be maintained.

Variation of the amplification as a function of frequency.

Variation of the limit level as a function of frequency.

Variation of the compression threshold and ratio in AGC (Automatic Gain Control) as a function of frequency.

Variation of attack and release times of AGC.

A combination of expansion and compression as a function of frequency.

Non-linear amplification as a function of frequency.

Frequency conversion upwards or downwards in frequency.

Recording of frequency changes in the signal,

for example formant transitions in speech sounds.

Variation of the balance of the microphone and pick-up-coil.

Of course it is also possible to implement other analog and/or digital signal processes. This is achieved thereby that a memory is arranged to store information/data for at least two unique signal process adjusted to different sound environments/listening situations and that a control unit, manual or automatic, is arranged to transmit information/data, for one of the unique signal processes, from the memory to the signal processor, to bring about one signal process adjusted to the particular sound environment/listening situation.

Brief Description of Drawings

The invention will be described in a preferred embodiment in the following text with reference to the attached drawings. Fig. 1 shows a block diagram of a signal processing device according to the invention and an external programming unit connected to it.

Fig. 2 shows a more detailed block diagram of the electronic circuits of the invention.

Description of the Preferred Embodiment

Figure 1 shows a signal processing device 1 according to the invention, and to which an external programming unit 2 can be connected via an input/output terminal 3. By means of the programming unit 2 information can be read in to, or out from, a memory 6. The signal processing device 1 consists mainly of a signal processor 4, a control unit 5, a memory 6, a microphone 7, an earphone 8 and a control gear 9, such as a switch, arranged to change the signal process of the signal processing device 1.

The signal processing device 1 is arranged thus that by manually activating the switch 9, or automatically by command from the signal processing unit 4, the control unit 5 transfers new information from the memory 6 to the signal processor 4 thereby specifying the signal process.

Fig. 2 shows a more detailed block diagram of the signal processing device 1. The signal processor 4 can be constructed with different techniques i.e. analog or digital signal processing, and with a variety of different signal processing systems. To clarify it is given one example of a signal processing system, which is based upon the principle that the input signal is split up in three frequency bands and each of the three signals is limited and attenuated. This signal processor 4 is based on analog technique all integrated on one chip using bipolar technology.

The control unit 5 and memory 6 are based on digital technique, all integrated in one chip using CMOS technology. The memory 6 is of non-volatile CMOS-type, in this case organized in 1×643 bits.

The signal processor 4 has two input terminals 10 and 11, and one input/output terminal 3. A microphone 7 is connected to input 10 and a tele-

or pick-up-coil 16 to input 11. The input/output terminal 3 is used as galvanic audio input or can be connected to an external programming unit 2 so that data can be written into the memory 6 or read out from the memory 6 to the programming unit 2.

A digitally controlled two-way switch 20a, which is controlled by the logic unit 21, is activated when data is transferred in or out.

The signal from the microphone 7 passes a capacitor 13a and is amplified 30 dB in the microphone amplifier 14a and then filtered in a high pass filter 15 ($f_c = 200$ Hz, 6 dB/octave). The signal from the pick-up-coil 16 is amplified 30 dB in the pick-up-coil amplifier 14b.

These two different signals are then attenuated (0—40 dB) in two digitally controlled attenuators 18a, 18b. The analog signals can also be electronically disconnected by the attenuators 18a, 18b. The attenuators 18a, 18b are each controlled by 8 bits words from the slave memory 27.

The signals from the microphone 7, the pick-up-coil 16 and the audio input 13 are added and amplified in the summing amplifier 22a and thereafter limited in a limiter 23a in order not to saturate a filter 24. The limiting is done with "soft" peak clipping utilizing the non-linearity properties of a diode.

The filter 24 is based on transconductance filters which provide a 4th order Butterworth filter and divides the signal in 3 channels; low-, band- and high-pass. The two crossover frequencies of the filter 24 are independently digitally controlled by two 8 bits words from the slave memory 27, in quarter of an octave steps 190—2,000 Hz and 500—6,000 Hz respectively.

The three output signals from the filter 24 (low-, band- and high-pass) are amplified in amplifiers 14c—14e, attenuated in attenuators 18c—18e and limited in limiters 23b—23d in the same fashion as mentioned earlier. In this way the level of limitation can be controlled digitally independently in each channel. Each of the three signals then pass through digitally controlled attenuators 18f—18h, where the signal levels in the different channels are set before they are added. After the summing amplifier 22b the signal passes a digitally controlled switch 20b with the purpose of avoiding disturbance when information is altered in the slave memory 27. After a volume control 26 the signal is amplified in an output amplifier 25, the output being connected to an earphone 8.

A triple averaging detector 19 is connected to each output of the amplifiers 14c—14e, in order to give signals to the logic unit 21. The purpose of this detector 19 is to cause new data to be automatically shifted into the slave memory 27, when suitable signals trigger the logic unit 21.

The slave memory 27 is a shift-register of 80 bits, which furnishes the above mentioned units with digital information.

The control unit 5 consists of a voltage doubler and regulator 36, a logic unit 21, which receives clock pulses from the voltage doubler and regu-

lator 36, a high voltage sensor 35, and a binary counter 34, which addresses the memory 6, and a digitally controlled switch 20c.

The memory 6 in this embodiment is organized in 1×643 bits, which means that the memory 6 can provide information for up to eight different listening situations, with 80 bits per listening situation. The three extra bits are used for the logic unit 21 to tell how many listening situations the hearing aid has been programmed for. It could be from two to eight different listening situations.

When the signal processor device 1 is turned on via the power switch 17, the voltage doubler and regulator 36 generates a power reset pulse to the logic unit 21 and the binary counter 34. Immediately after the reset pulse the logic unit 21 operates in the following manner:

- Generates a pulse to the switch 20b, connecting poles 1 and 2, during data transfer.

- Sets the memory 6 in read mode during transfer of data.

- Generates eighty-three clockpulses to the counter 34. The three first bits are transferred to the logic unit 21. The remaining eighty bits of data from the memory 6 are transferred to the slave memory 27.

- Generates eighty clock pulses synchronously to the slave memory 27.

The signal processing device 1 is now operating for the first listening situation.

When the hearing aid wearer wants to change the signal processing device 1 for another listening situation he pushes the manual switch 9, which triggers the logic unit 21 and operates in the following manner:

- Generates a pulse to the switch 20b, connecting poles 1 and 2, during data transfer.

- Addresses the memory 6 for new location of eighty new bits of information.

- Sets the memory 6 in read mode during data transfer.

- Generates eighty clockpulses to the counter 34. Eighty bits of data from the memory 6 are transferred to the slave memory 27.

- Generates eighty clock pulses synchronously to the slave memory 27.

The signal processing device 1 now operates for the second listening situation. If the hearing aid wearer again pushes the manual switch 9, the process is repeated and the hearing aid operates for a third listening situation.

When the user activates the manual switch 9, and the aid is operating for the last pre-programmed listening situation, as indicated by the above mentioned first three bits, the logic unit 21 again transfers the data for the first listening situation to the slave memory 27. In this way the data information of the different listening modes are transferred to the slave memory 27 in a cyclic manner.

If the hearing aid wearer does not know for which listening mode the hearing aid operates for the moment he turns the aid off and on with the power switch 17 and the hearing aid will operate

for the first listening situation.

The control unit 5 can also transfer data automatically to the slave memory 27, if the hearing aid wearer moves from one acoustical listening situation to another. A suitable change in the information from the triple averaging detector 19 triggers the logic unit and new data information is transferred from the memory 6 to the slave memory 27, for that particular listening situation.

When data is written to the memory 6 from an external programming unit 2 or data is read out from the memory 6 to the external programming unit 2, the battery 33 is removed, and a three pole adaptor (not shown in figure) from the programming unit 2 is connected to the battery connectors 28, 29 and to the data input/output 3.

Programming of the memory 6 is always first accomplished by an erase pulse and then all the 643 bits are transferred in series to the memory 6. This is done by raising the voltage to the connector 28 and pulsing it with about 1 kHz and synchronously transferring data from the programming unit 2 via the connector 3 to the memory 6.

The logic unit 21 operates in the following manner when it receives a pulse longer than $200 \mu s$ from the high voltage sensor 35.

- Generates a pulse to the switches 20a, 20b and 20c, connecting poles 1 and 2, during data transfer.

- Sets the memory 6 in erase mode. The total memory area is now erased by the first high voltage pulse about 1 ms long.

- Sets the memory 6 in write mode, during data transfer.

- Each pulse from the high voltage sensor 35 advances the address word of the memory 6 by one bit, via the logic unit 21 and the counter 34.

With the high voltage pulses, about 1 ms long, to the memory 6, and with data coming synchronously from the programming unit 2 via terminal 3, switches 20a and 20c, the memory 6 is being programmed.

To transfer data from the memory 6 to the programming unit 2, the logic unit 21 is triggered via the high voltage sensor 35, with one very short high voltage pulse less than $50 \mu s$. The programming unit 2 first generates a pulse to the terminal 3 for incrementing the address word for the memory 6 and then reads the first data bit from the memory 6, again generates a pulse and reads out the next data bit and so on, until all 643 bits are read out in series from the memory 6 to the programming unit 2.

The logic unit 21 operates in the following manner:

- Generates a pulse to the switches 20a, 20b and 20c, connecting poles 1 and 2, during data transfer.

- Sets the memory 6 in read mode during data transfer.

- Each incrementing pulse from the programming unit 2 increments the address word for the memory 6 by one bit via the logic unit 21 and the counter 34.

In this manner all data (643 bits) from the

memory 6 is transferred to the programming unit 2, via the switches 20c, 20a and terminal 3.

The invention is of course not limited to the above disclosed embodiment. A number of alternative embodiments are possible within the scope of the claims. Therefore it is possible to use the invention for example in a number of different applications where it is necessary that some signal process automatically or manually should be changed in the signal processing device, when the sound environment or the listening situation is changed. The electronic components can also of course be of different kinds. For example the memory 6 may be of either a volatile or a non-volatile type.

Claims

1. Programmable signal processing device (1), mainly intended for persons having impaired hearing, and of the kind which process an input signal containing information such as speech or music and which includes an electronically controlled signal processor (4) characterized in that a memory (6) is arranged to store information/data for at least two unique signal processes adjusted to different sound environments/listening situations and that a control unit (5) manual or automatic is arranged to transmit information/data, for one of the unique signal processes, from the memory (6) to the signal processor (4), to bring about one signal process adjusted to a particular sound environment/listening situation.

2. Programmable signal processing device (1) according to claim 1, characterized in that a control gear (9) is arranged to influence the control unit (5), manually, thus that digital information is transmitted from the memory (6) to the signal processor (4) for specifying the signal process.

3. Programmable signal processing device (1) according to claim 1 or 2, characterized in that the signal processor (4) is arranged to influence the control unit (5) automatically, depending on the sound environment, thus that digital information is transmitted from the memory (6) to the signal processor (4) for specifying the signal process.

4. Programmable signal processing device (1) according to one or more of the preceding claims, characterized in that a programming unit (2) is connectable to an input/output terminal (3) of the signal processing device (1) and arranged to influence the control unit (5) thus that digital information is transmitted between the programming unit (2) and the memory (6).

5. Programmable signal processing device (1) according to one or more of the preceding claims, characterized in that two attenuators (18a, 18b) and one switch (20a) are connected to input terminals of a summing amplifier (22a) and are arranged to balance and adjust the signal levels supplied to input terminals (3, 10, 11) of the signal processing device (1) from different signal sources, to the actual sound environment/listening situation.

Reindications

1. Système de traitement de signaux programmable (1), principalement destiné à des personnes ayant une ouïe altérée et du type qui traite un signal d'entrée contenant des informations telles que des paroles ou de la musique et qui comprend une unité de traitement de signaux ou processeur commandé électroniquement (4), caractérisé en ce qu'une mémoire (6) est agencée pour mémoriser des informations/données pour au moins deux traitements de signaux uniques réglés pour différents environnements sonores/conditions d'écoute et en ce qu'une unité de commande (5) manuelle ou automatique est agencée pour transmettre des informations/données, pour un des traitements de signaux uniques, de la mémoire (6) au processeur (4), pour faire en sorte qu'un traitement de signaux soit réglé pour un environnement sonore/condition d'écoute particulier.

2. Système de traitement de signaux programmable (1) selon la revendication 1, caractérisé en ce qu'un dispositif de commande (9) est agencé pour influencer sur l'unité de commande (5), manuellement, pour que des informations numériques soient transférées de la mémoire (6) jusqu'au processeur (4) pour spécifier le traitement de signaux.

3. Système de traitement de signaux programmable (1) selon l'une quelconque des revendications 1 et 2, caractérisé en ce que le processeur (4) est agencé pour influencer sur l'unité de commande (5) automatiquement; en fonction de l'environnement sonore, pour que des informations numériques soient transférées de la mémoire (6) jusqu'au processeur (4) pour spécifier le traitement de signaux.

4. Système de traitement de signaux programmable (1) selon l'une quelconque des revendications 1 à 3, caractérisé en ce qu'une unité de programmation (2) peut être connectée à une borne d'entrée/sortie (3) du système de traitement de signaux (1) et en ce qu'elle est agencée pour influencer sur l'unité de commande (5) pour que des informations numériques soient transmises entre l'unité de programmation (2) et la mémoire (6).

5. Système de traitement de signaux programmable (1) selon l'une quelconque des revendications 1 à 4, caractérisé en ce que deux atténuateurs (18a, 18b) et un commutateur (20a) sont connectés à des bornes d'entrée d'un amplificateur de sommation (22a) et en ce qu'ils sont agencés pour équilibrer et régler les niveaux de signaux envoyés à des bornes d'entrée (3, 10, 11) du système de traitement de signaux (1) provenant de différentes sources de signaux pour le présent environnement sonore/condition d'écoute.

Patentansprüche

1. Programmierbare Signalverarbeitungseinrichtung (1) hauptsächlich für Personen mit

herabgesetztem Gehör, welche ein Eingangssignal, das solche Informationen wie Reden oder Musik enthält, verarbeitet und welche einen elektronisch gesteuerten Prozessor (4) umfasst, dadurch gekennzeichnet, dass ein Speicher (6) vorgesehen ist, die Informationen bzw. Daten von mindestens zwei eindeutigen an verschiedenen Schallumgebungen oder Hörsituationen angepasste Signalbehandlungsverfahren zu speichern, und dass eine Steuereinheit (5) angeordnet ist bei manueller oder automatischer Betätigung Informationen oder Daten für eine der eindeutigen Signalbehandlungsverfahren von dem Speicher (6) zum Signalprozessor (4) zu überführen um ein für eine besondere Schallumgebung oder Hörsituation angepasste Signalbehandlungsverfahren zu erreichen.

2. Programmierbare Signalverarbeitungseinrichtung nach Anspruch 1, dadurch gekennzeichnet, dass ein Manöverorgan (9) vorgesehen ist, bei manueller Aktivierung die Steuereinheit (5) so zu beeinflussen, dass digital gespeicherte Information vom Speicher (6a, 6b) zum Signalprozessor (4) zur Änderung des Signalbehandlungsverfahrens überführt wird.

3. Programmierbare Signalverarbeitungseinrichtung nach Anspruch 1 oder 2, dadurch

gekennzeichnet, dass der Signalprozessor (4) angeordnet ist, die Steuereinheit (5) in Abhängigkeit der Schallumgebung automatisch so zu beeinflussen, dass digital gespeicherte Information vom Speicher (6a, 6b) zum Signalprozessor (4) zur Änderung des Signalbehandlungsverfahrens überführt wird.

4. Programmierbare Signalverarbeitungseinrichtung nach einem oder mehreren der obigen Ansprüche, dadurch gekennzeichnet, dass eine Programmierungseinheit (2) an den Ein-/Ausgang (3) der Signalverarbeitungseinrichtung (1) angeschlossen und vorgesehen ist, die Steuereinheit (5) so zu beeinflussen, dass digital kodifizierte Information zwischen der Programmierungseinheit (2) und dem Speicher (6a, 6b) überführt wird.

5. Programmierbare Signalverarbeitungseinrichtung nach einem oder mehreren der obigen Ansprüche, dadurch gekennzeichnet, dass zwei Regelglieder (18, 18b) und ein Schalter (20a) an den Eingang eines Summierungsverstärkers (22a) angeschlossen und vorgesehen sind, die zu den Eingängen (3, 10, 11) von verschiedenen Signalquellen zugeführten Signalebenen an die aktuelle Schallumgebung oder Hörsituation anzugleichen und anzupassen.

5

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FIG. 1

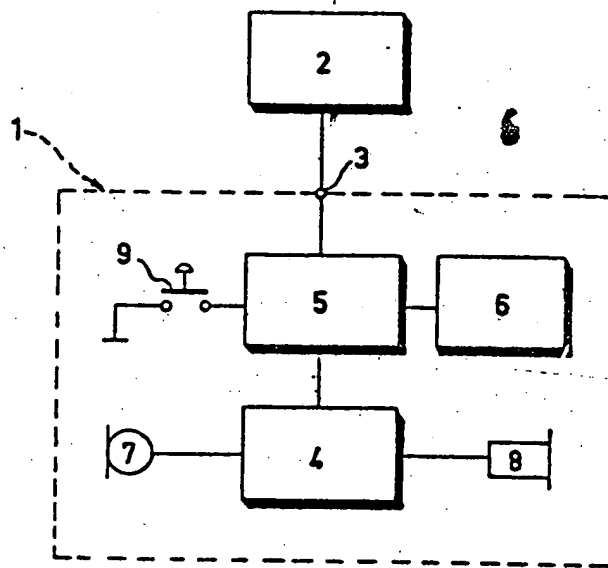


FIG. 2

